IN THE UNITED STATES
PATENT AND TRADEMARK OFFICE

PATENT APPLICATION

Appellants:	**Hassan Hagirahim et al.**		
Case:	**Hagirahim 8-7 (LCNT/121667)**		
Serial No.:	**09/659,650**	Filed:	**September 12, 2000**
Examiner:	**Richard Chang**	Group Art Unit:	**2616**

Title: **METHOD AND APPARATUS FOR PROVIDING EFFICIENT VoIP GATEWAY-TO-GATEWAY COMMUNICATION**

Confirmation #: **6967**

MAIL STOP APPEAL BRIEF-PATENTS
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

APPEAL BRIEF

Appellants submit this Appeal Brief to the Board of Patent Appeals and Interferences on appeal from the decision of the Examiner of Group Art Unit 2616 mailed July 25, 2006, finally rejecting claims 1, 2, 6-14, 18-28, 32 and 33. The final rejection of claims 1, 2, 6-14, 18-28, 32 and 33 is appealed.

In the event that an extension of time is required for this Appeal Brief to be considered timely, and a petition therefor does not otherwise accompany this response, any necessary extension of time is hereby petitioned for.

The Commissioner is authorized to charge the $500 Appeal Brief filing fee, and any additional fees required to make this Appeal Brief timely and acceptable to the Office, to counsel's Deposit Account No. 20-0782/LCNT/121667.

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Real Party in Interest

The present application has been assigned to Lucent Technologies Inc. of Murray Hill, New Jersey.

Related Appeals and Interferences

Appellants assert that no other appeals or interferences are known to Appellants, Appellants' legal representative, or assignee, which will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

Status of Claims

Claims 1, 2, 6-14, 18-28, 32 and 33 are pending in the application. Claims 1-32 were originally filed in the application; claims 33-34 were added by amendment; claims 1-2, 6-7, 13, 18, 25, 27, 32 and 33 have been amended; claims 3-5, 15-17, 29-31 and 34 have been canceled. Claims 1, 2, 6-14, 18-28, 32 and 33 stand finally rejected as discussed below. The final rejections of claims 1, 2, 6-14, 18-28, 32 and 33 are appealed. The pending claims are shown in the attached Claims Appendix. Since amendments submitted in response to the final rejection mailed July 25, 2006 have not been entered, claims discussed herein are the claims from the response to the non-final office action mailed February 16, 2006.

Status of Amendments

Amendments submitted in response to the final rejection mailed July 25, 2006 have not been entered. Appellants submit that the amendments to the independent claims submitted in response to the final rejection mailed July 25, 2006 were merely submitted in order to correct a typographical error, and to correct a potential clarity issue. All other amendments have been entered.

Summary of Claimed Subject Matter

The embodiments of the present invention are generally directed to providing efficient Voice-over-Internet-Protocol (VoIP) gateway-to-gateway communications. The present invention advantageously provides a means of communicating voice traffic between VoIP gateways in composite call formation form without requiring a conversion of the voice traffic payloads into a different format between gateways. Since the voice traffic payloads remain intact, signal degradation and delay which result from converting payloads into different formats is avoided. In this manner, the present invention provides a signal with reduced overhead where at least two conversations are transmitted between VoIP gateways, thereby providing a substantial improvement over previous VoIP gateway-to-gateway communication.

A method according to one embodiment of the present invention includes receiving first voice traffic at a first VoIP gateway, determining whether a destination is serviced by a second VoIP gateway, multiplexing the first voice traffic with second voice traffic at the first VoIP gateway if the second voice traffic is being provided to the second VoIP gateway and transporting the multiplexed voice traffic to the second VoIP gateway utilizing a plurality of transport packets responsive to an affirmative determination that the destination is serviced by the second VoIP gateway. The transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets. The UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet, where a modified RTP packet comprises at least one of: a Payload field for containing a voice traffic, a Call Identifier field for identifying a caller, a Length Indicator field for identifying the size of the payload field, and a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

An apparatus according to one embodiment of the present invention includes a first Voice over Internet Protocol (VoIP) gateway for receiving first voice traffic, determining whether the destination is serviced by a second VoIP gateway, multiplexing the first voice traffic with second voice traffic if the second voice traffic is being provided to the second VoIP gateway, and transporting the multiplexed voice traffic to the second VoIP gateway utilizing a plurality of transport packets responsive to an affirmative determination that the destination is serviced by the second VoIP gateway. The

transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets. The UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet, where a modified RTP packet comprises at least one of: a Payload field for containing a voice traffic, a Call Identifier field for identifying a caller, a Length Indicator field for identifying the size of the payload field, and a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

An apparatus according to one embodiment of the present invention includes a processor and a storage device coupled to the processor and including instructions for controlling the processor, where the processor is operative with the instructions to receive first voice traffic at a first VoIP gateway, determine whether the destination is serviced by a second VoIP gateway, multiplex the first voice traffic with second voice traffic at the first VoIP gateway if the second voice traffic is being provided to the second VoIP gateway, and transport the multiplexed voice traffic to the second VoIP gateway utilizing a plurality of transport packets responsive to an affirmative determination that the destination is serviced by the second VoIP gateway. The transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets. The UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet, where a modified RTP packet comprises at least one of: a Payload field for containing a voice traffic, a Call Identifier field for identifying a caller, a Length Indicator field for identifying the size of the payload field, and a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

An apparatus according to one embodiment of the present invention includes means for receiving first voice traffic at a first VoIP gateway, means for determining whether the destination is serviced by a second VoIP gateway, means for multiplexing the first voice traffic with second voice traffic at the first VoIP gateway if the second voice traffic is being provided to the second VoIP gateway, and means for transporting the multiplexed voice traffic to the second VoIP gateway utilizing a plurality of transport packets responsive to an affirmative determination that the destination is serviced by the second VoIP gateway. The transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets. The UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet, where a modified RTP packet

comprises at least one of: a Payload field for containing a voice traffic, a Call Identifier field for identifying a caller, a Length Indicator field for identifying the size of the payload field, and a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

A method according to one embodiment of the present invention includes receiving voice traffic at a first Voice over Internet Protocol (VoIP) gateway and transporting the voice traffic to a second VoIP gateway utilizing a plurality of transport packets if a destination of the voice traffic is serviced by the second VoIP gateway. The transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets. The UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet, where a modified RTP packet comprises at least one of: a Payload field for containing a voice traffic, a Call Identifier field for identifying a caller, a Length Indicator field for identifying the size of the payload field, and a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

For the convenience of the Board of Patent Appeals and Interferences, Appellants' independent claims are presented below in claim format with reference numerals corresponding to the figures and appropriate citations to at least one portion of the specification for each element of the appealed claims.

Claim 1 positively recites (with reference numerals and cites to Appellants' specification added, where applicable):

1. (Previously presented) A method, comprising the steps of:

receiving (304) a first voice traffic at a first Voice over Internet Protocol (VoIP) gateway (122); (Pg. 4, Lines 16-17; Pg. 6, Lines 14-16)

determining (306) whether a destination (132, 134, 136, 138) is serviced by a second VoIP gateway (128); (Pg. 4, Lines 19-22; Pg. 6, Lines 17-27)

multiplexing (316, 322), at said first VoIP gateway (122), said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway (128); and (Pg. 4, Lines 20-26; Pg. 7, Lines 5-7, 20-21)

transporting (316, 320, 322) said multiplexed voice traffic to said second VoIP gateway (128) utilizing a plurality of transport packets, responsive to an affirmative

determination that said destination (132, 134, 136, 138) is serviced by said second VoIP gateway (306), wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet (400, 500), (Pg. 4, Lines 20-27; Pg. 7, Lines 12-24)

wherein said modified RTP packet (400, 500) comprises at least one of:

a Payload (506, 510, 514) field for containing a voice traffic;

a Call Identifier field (404) for identifying a caller;

a Length Indicator field (406) for identifying the size of the payload field; and

a Header Error Check field (408) for identifying errors in the Call Identifier field and the Length Indicator field (Pg. 8, Lines 19-27; Pg. 9, Lines 3-12).

Claim 13 positively recites (with reference numerals and cites to Appellants' specification added, where applicable):

13. (Previously presented) In a communication system (100) for transporting voice traffic over an Internet Protocol (IP) network (126) to a destination (132, 134, 136, 138), apparatus comprising:

a first Voice over Internet Protocol (VoIP) gateway (122), for receiving (304) a first voice traffic; (Pg. 4, Lines 16-17; Pg. 6, Lines 14-16)

said first VoIP gateway (122) determining (306) whether said destination (132, 134, 136, 138) is serviced by a second VoIP gateway (128); (Pg. 4, Lines 19-22; Pg. 6, Lines 17-27)

said first VoIP gateway (122) multiplexing (316, 322) said first voice traffic with a second voice traffic, if said second voice traffic is being provided to said second VoIP gateway (128); (Pg. 4, Lines 20-26; Pg. 7, Lines 5-7, 20-21)

said first VoIP gateway (122) transporting (316, 320, 322) said multiplexed voice traffic to said second VoIP gateway (128) utilizing a plurality of transport packets, responsive to an affirmative determination that said destination (132, 134, 136, 138) is serviced by said second VoIP gateway (128), wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP

packets transport at least one modified Real-time Transport Packet (RTP) packet (400, 500), (Pg. 4, Lines 20-27; Pg. 7, Lines 12-24)

wherein said modified RTP packet (400, 500) comprises at least one of:

a Payload field (506, 510, 514) for containing a voice traffic;

a Call Identifier field (404) for identifying a caller;

a Length Indicator field (406) for identifying the size of the payload field; and

a Header Error Check field (408) for identifying errors in the Call Identifier field and the Length Indicator field. (Pg. 8, Lines 19-27; Pg. 9, Lines 3-12)

Claim 25 positively recites (with reference numerals and cites to Appellants' specification added, where applicable):

25. (Previously presented) A Voice over Internet Protocol (VoIP) gateway (122) for transporting voice traffic over an Internet Protocol (IP) network (126) to a destination (132, 134, 136, 138), comprising:

a processor (220); and

a storage device (230) coupled to said processor (220) and including instructions (300) for controlling said processor (220), said processor (220) operative with said instructions (300) to:

receive (304) a first voice traffic at said VoIP gateway (122); (Pg. 4, Lines 16-17; Pg. 6, Lines 14-16)

determine (306) whether said destination (132, 134, 136, 138) is serviced by a second VoIP gateway (128); (Pg. 4, Lines 19-22; Pg. 6, Lines 17-27)

multiplex (316, 322), at said VoIP gateway (122), said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway (128); and (Pg. 4, Lines 20-26; Pg. 7, Lines 5-7, 20-21)

transport (316, 320, 322) said multiplexed voice traffic to said second VoIP gateway (128) utilizing a plurality of transport packets, responsive to an affirmative determination that said destination (132, 134, 136, 138) is serviced by said second VoIP gateway (128), wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at

least one modified Real-time Transport Packet (RTP) packet (400, 500), (Pg. 4, Lines 20-27; Pg. 7, Lines 12-24)

wherein said modified RTP packet (400, 500) comprises at least one of:

a Payload field (506, 510, 514) for containing a voice traffic;

a Call Identifier field (404) for identifying a caller;

a Length Indicator field (406) for identifying the size of the payload field; and

a Header Error Check field (408) for identifying errors in the Call Identifier field and the Length Indicator field. (Pg. 8, Lines 19-27; Pg. 9, Lines 3-12)

Claim 27 positively recites (with reference numerals and cites to Appellants' specification added, where applicable):

27. (Previously presented) A Voice over Internet Protocol (VoIP) gateway (122), for transporting voice over an Internet Protocol (IP) network (126), to a destination (132, 134, 136, 138), comprising:

means for receiving (210) a first voice traffic at said VoIP gateway (122); (Pg. 4, Lines 16-17; Pg. 6, Lines 14-16)

means for determining (220) whether said destination (132, 134, 136, 138) is serviced by a second VoIP gateway (128); (Pg. 4, Lines 19-22; Pg. 6, Lines 17-27)

means for multiplexing (220, 230, 240), at said VoIP gateway (122), said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway (128); and (Pg. 4, Lines 20-26)

means for transporting (210) said multiplexed voice traffic to said second VoIP gateway (128) utilizing a plurality of transport packets, responsive to an affirmative determination that said destination (132, 134, 136, 138) is serviced by said second VoIP gateway (128), wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet (400, 500), (Pg. 4, Lines 20-27; Pg. 7, Lines 12-24)

wherein said modified RTP packets (400, 500) comprise at least one of:

a Payload field (506, 510, 514) for containing a voice traffic;

a Call Identifier field (404) for identifying a caller;

a Length Indicator field (406) for identifying the size of the payload field; and

a Header Error Check field (408) for identifying errors in the Call Identifier field and the Length Indicator field. (Pg. 8, Lines 19-27; Pg. 9, Lines 3-12)

Claim 33 positively recites (with reference numerals and cites to Appellants' specification added, where applicable):

33. (Previously presented) A method, comprising the steps of:

receiving (304) a voice traffic at a first Voice over Internet Protocol (VoIP) gateway (122); (Pg. 4, Lines 16-17; Pg. 6, Lines 14-16)

transporting (316, 320, 322) the voice traffic to a second VoIP gateway (128) utilizing a plurality of transport packets if a destination (132, 134, 136, 138) of the voice traffic is serviced by the second VoIP gateway (128) (Pg. 6, Lines 17-27), wherein the transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet (400, 500), (Pg. 4, Lines 20-27; Pg. 7, Lines 12-24)

wherein said modified RTP packets (400, 500) comprise at least one of:

a Payload field (506, 510, 514) for containing a voice traffic;

a Call Identifier field (404) for identifying a caller;

a Length Indicator field (406) for identifying the size of the payload field; and

a Header Error Check field (408) for identifying errors in the Call Identifier field and the Length Indicator field. (Pg. 8, Lines 19-27; Pg. 9, Lines 3-12)

Grounds of Rejection to be Reviewed on Appeal

The Examiner has rejected claims 1-2, 13-14, 27-28 and 33 under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent 6,363,065 to Thornton et al. (hereinafter "Thornton") in view of U.S. Patent 6,918,034 to Sengodan et al. (hereinafter "Sengodan") and further in view of U.S. Patent 6,717,948 to Subbiah (hereinafter "Subbiah").

The Examiner has rejected claims 6-12, 18-26, and 32 under 35 U.S.C. §103(a) as being unpatentable over Thornton in view of Sengodan and Subbiah and further in view of U.S. Patent 5,600,653 to Chitre et al. (hereinafter "Chitre").

Arguments

35 U.S.C. §103(a) Rejection of Claims 1-2, 13-14, 27-28 and 33

The Examiner has rejected claims 1-2, 13-14, 27-28 and 33 under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent 6,363,065 to Thornton et al. (hereinafter "Thornton") in view of U.S. Patent 6,918,034 to Sengodan et al. (hereinafter "Sengodan") and further in view of U.S. Patent 6,717,948 to Subbiah (hereinafter "Subbiah").

Claims 1-2:

In general, Thornton discloses methods and apparatus for a telephony gateway intended for use at opposite ends of a data network connection. At each end of the connection, telephony gateways are used in conjunction with a private branch exchange (PBX) for automatically routing calls, e.g., voice, data, and facsimile, between two peer PBXs over either a public switched telephone network (PSTN) or a data network. The routing is based on, among other things cost considerations for handling each such call and called directory numbers, monitoring quality of service provided through the data network, and switching such calls back and forth between the PSTN and data network, as needed, in response to dynamic changes in the quality of service, such that the call is carried over a connection then providing a sufficient quality of service. (Thornton, Abstract).

Thornton, however, fails to teach or suggest Appellants' claim 1, as a whole. Namely, Thornton fails to teach or suggest at least the limitations of "determining whether a destination is serviced by a second VoIP gateway, multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 1.

In the Final Office Action mailed July 25, 2006, the Examiner asserts that Thornton teaches Appellants' limitation of "determining whether a destination is serviced

by a second VoIP gateway," as claimed in Appellants' claim 1. Specifically, in the Final Office Action mailed July 25, 2006, the Examiner states that "...the originating VoIP gateway does affirmatively determine that destination VoIP gateway is present for connection as the necessary condition for a connection between the two VoIP gateway[s]." (Final Office Action, July 25, 2006, Pg. 2, Emphasis added). In other words, in rejecting Appellants' claim 1, the Examiner concludes that Thornton teaches determining that a second VoIP gateway is present in order to support a connection between a first VoIP gateway and the second VoIP gateway.

Appellants respectfully submit, however, that this is not what is claimed in Appellants' claim 1. Rather, Appellants' limitation is directed to the association between a second VoIP gateway and a destination, not to the association between the second VoIP gateway and first VoIP gateway. Specifically, Appellants' limitation specifies a determination as to whether a destination is serviced by a second VoIP gateway. By contrast, Thornton merely teaches generic peering between a first VoIP gateway and a second VoIP gateway in order to support communication between the VoIP gateways. A general statement that a first VoIP gateway is peered to a second VoIP gateway so as to provide communication between the VoIP gateways, as taught in Thornton, simply does not teach or suggest "determining whether a destination is serviced by a second VoIP gateway," as claimed in Appellants' claim 1. Thornton is devoid of any teaching or suggestion of any such determination as to whether a destination is serviced by a VoIP gateway.

Furthermore, with respect to the cited portion of the Final Office Action mailed July 25, 2006, by stating "destination VoIP gateway", and further stating "determine that destination VoIP gateway is present for connection" in the cited portion of the Final Office Action, the Examiner seems to be asserting that one of the peer-to-peer gateways of Thornton teaches the "destination" of Appellants' claim 1. Appellants' claim 1, however, clearly recites a first VoIP gateway, a second VoIP gateway, and a destination. Thus, Appellants' claim 1 clearly indicates that the destination is not the first VoIP gateway or the second VoIP gateway. Therefore, the Examiner's reading of Appellants' limitation of "determining whether a destination is serviced by a second VoIP gateway" is incorrect.

In the Final Office Action mailed July 25, 2006, the Examiner cites a specific portion of Thornton (Col. 13, Lines 57-62; Col. 14, Lines 3-8), asserting that the cited portion of Thornton teaches Appellants' limitation of "determining whether a destination is serviced by a second VoIP gateway," as claimed in Appellants' claim 1. The cited portion of Thornton, however, merely states that a digitized telephony signal is converted into IP packets which are transmitted over a data network to a peer gateway using IP addresses, and that each called number has an associated IP address known to both peer gateways. Specifically, the cited portions of Thornton state:

> "[The] DSP and the microcontroller convert the digitized telephony signal for that call into suitable IP packets and transmit those packets, with appropriate IP addresses, over the LAN for subsequent carriage over the data network to a <u>peer gateway</u>."
> (Thornton, Col. 13, Lines 57-62, Emphasis added).

> "<u>Each separate called number has an associated IP address, which ultimately is known to both peer gateways ...</u> such that the peered gateways can properly address the IP packets to their unique called destination."
> (Thornton, Col. 14, Lines 3-8, Emphasis added).

The cited portions of Thornton fail to teach or suggest any determining step, much less <u>determining whether</u> <u>a destination is serviced by a VoIP gateway</u>, as claimed in Appellants' claim 1. Rather, the above-quoted portions of Thornton clearly evidence two critical facts; namely, (1) that a peer to peer communication is envisioned (i.e., that each call is definitely being made from a first gateway to a second gateway, where both gateways have substantially equivalent topologies), and (2) that both gateways have full, *a priori*, knowledge of the IP addresses of other gateways. Thus, not only does Thornton fail to teach or suggest Appellant's limitation of "determining whether a destination is serviced by a second VoIP gateway," but, since the Thornton arrangement is specifically directed to a peer-to-peer arrangement in which each IP address is already known to <u>both peer gateways</u>, there is simply <u>no need within the Thornton arrangement to perform the step of "determining whether a destination is serviced by a second VoIP gateway</u>."

In the Final Office Action mailed July 25, 2006, the Examiner further cites another specific portion of Thornton (Col. 4, Line 65 – Col. 5, Line 8), asserting that the cited

portion of Thornton teaches Appellants' limitation of "determining whether a destination is serviced by a second VoIP gateway," as claimed in Appellants' claim 1. The cited portion of Thornton, however, merely states that a gateway determines network quality using dynamic quality measurements and that if either of the peer-to-peer gateways involved in a call determines that network quality has either increased or decreased to necessitate an auto-switch between a data network and a PSTN. Specifically, the cited portion of Thornton states:

> "In particular, the inventive gateway determines network quality through dynamic measurements of latency, packet loss and error rate (jitter). Should either gateway involved in a call determine that network quality has either increased or decreased to necessitate an auto-switch either to the data network from the PSTN or the opposite, that gateway (hereinafter, for simplicity of reference, the "calling gateway") will initiate an information exchange, using our inventive extensions to the H.323 protocol with its peer gateway (hereinafter, the "called" gateway)."
> (Thornton, Col. 4, Line 65 – Col. 5, Line 8, Emphasis added).

The cited portion of Thornton fails to teach or suggest determining whether a destination is serviced by a VoIP gateway, as claimed in Appellants' claim 1. Rather, the cited portion of Thornton teaches a network quality determination in which the quality provided to a call is determined by measuring latency, packet loss, and other quality measurements. As stated in the cited portion of Thornton, each gateway switches a call between a data network and a PSTN in response to determining a change in the network quality. A determination of network quality using dynamic network quality measurements, as taught in Thornton, does not teach or suggest determining whether a destination is serviced by a VoIP gateway, as claimed in Appellants' claim 1.

As such, for at least the reasons discussed hereinabove, Thornton fails to teach or suggest the limitation of "determining whether a destination is serviced by a second VoIP gateway," as claimed in Appellants' claim 1.

Furthermore, Thornton also fails to teach or suggest Appellants' limitation of "multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway," as claimed in Appellants' claim 1.

In the Final Office Action mailed July 25, 2006, the Examiner cites a specific portion of Thornton for teaching Appellants' limitation of "multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway." The cited portion of Thornton, however, actually states that, for transmission of voice data over the data network rather than the PSTN, multiplexing is <u>not</u> performed. Rather, Thornton teaches that, instead of multiplexing, voice traffic is instead sent to a DSP and microcontroller within the gateway which convert the voice traffic into IP packets for transmission over the data network. Specifically, the cited portion of Thornton states:

> "Alternatively, <u>if the gateway were to route an outgoing telephony call from a calling device</u>, such as a telephone, computer modem or facsimile machine, connected to the PBX <u>over the private data network</u> (to effectuate a "Voice over IP" or <u>VoIP call</u>) <u>instead of the PSTN</u>, TDM switch 250, based on control information provided by microcontroller 240, <u>connects an incoming time slot for that call</u>, <u>not</u> <u>to a time slot via T1/E1 transceiver/framer 2 and, from there, to an outgoing T1 trunk</u>, <u>but rather</u>, via TDM bus 228, <u>to an input of a DSP</u> then available within DSPs 225 <u>and ultimately to microcontroller 240</u>. Collectively, <u>that DSP and the microcontroller convert the digitized telephony signal for that call into suitable IP packets</u> and transmit those packets, with appropriate IP addresses, over the LAN for subsequent carriage over the data network to a peer gateway."
> (Thornton, Col. 13, Lines 48 – 62, Emphasis added).

Thus, although Thornton teaches multiplexing of voice traffic, Thornton merely teaches that multiplexing is performed for transmission of voice traffic over a PSTN. For example, Thornton teaches that "...a signal on a channel in an incoming T1 trunk, such as that carried by TDM lines 268, and originating from the PSTN, can be switched, through switch 250, to a corresponding time slot on an outgoing T1 trunk, such as over TDM lines 278, to the PBX, and vice versa, in order to support carriage of that call over the PSTN between caller and called locations." (Thornton, Col. 13, Lines 35-41). In other words, Thornton fails to teach or suggest multiplexing voice traffic for transporting the multiplexed voice traffic to a second VoIP gateway, as claimed in Appellants' claim 1.

Furthermore, since Thornton fails to teach or suggest limitations of "determining whether a destination is serviced by a second VoIP gateway" and "multiplexing, at said

first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway," Thornton must also fail to teach or suggest "transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, <u>responsive to an affirmative determination that said destination is serviced by said second VoIP gateway</u>," as claimed in Appellants' claim 1. Again, Thornton does not require such a determination since Thornton provides a peer-to-peer gateway architecture in which each IP address is already known to <u>both peer gateways</u>.

As such, Thornton fails to teach or suggest Appellants' claim 1, as a whole.

Furthermore, Sengodan and Subbiah fail to bridge the substantial gap between Thornton and Appellants' claim 1. Namely, Sengodan and Subbiah, alone or in combination with each other and Thornton, fail to teach or suggest at least the limitations of "determining whether a destination is serviced by a second VoIP gateway, multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway, and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 1.

In general, Sengodan is directed toward encryption and authentication of mini-packets in a multiplexed real time protocol (RTP) payload. As taught in Sengodan, mini-packets are added to the RTP payload, which is then padded to ensure that each mini-packet is an integral multiple of a predetermined block size. The disclosed arrangement is utilized within the context of a VoIP system in which each user sharing a single RTP/UDP/IP connection is associated with a respective channel identifier (CID). (Sengodan, Abstract).

Sengodan, however, alone or in combination with Thornton and/or Subbiah, fails to teach or suggest Appellants' claim 1, as a whole. Namely, Sengodan fails to teach or suggest determining whether a destination is serviced by a second VoIP gateway, multiplexing first voice traffic with second voice traffic if the second voice traffic is being provided to the second VoIP gateway, and transporting multiplexed voice traffic to the second VoIP gateway utilizing a plurality of transport packets in response to an

affirmative determination that the destination is serviced by the second VoIP gateway, as claimed in Appellants' claim 1.

In the Final Office Action mailed July 25, 2006, the Examiner cites a portion of Sengodan for teaching limitations of Appellants' invention. The portion of Sengodan cited by the Examiner, however, merely describes the assembly of RTP/UDP/IP packets from mini-packets. Specifically, the cited portion of Sengodan states that "...assembly of mini-packets into a single RTP/UDP/IP payload 300 is shown in FIG. 3. The mini-packets 330, 350, 370 follow the IP header 310, the UDP header 312 and the RTP header 314. Each mini-packet 330, 350, 370 is delineated by two byte mini-headers 320, 340, 360, respectively. This approach requires a simple de-multiplexing algorithm at a receiver." (Sengodan, Col. 7, Lines 46-52).

In other words, the cited portion of Sengodan merely describes the format of an RTP packet having mini-packets. Sengodan fails to teach or suggest Appellants' limitation of "determining whether a destination is serviced by a second VoIP gateway," as claimed in Appellants' claim 1. Furthermore, although Sengodan states that a de-multiplexing algorithm is required at the receiver, such a general statement does not teach or even suggest multiplexing first voice traffic with second voice traffic if the second voice traffic is being provided to the second VoIP gateway or transporting multiplexed voice traffic to a second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that the destination is serviced by the second VoIP gateway, as claimed in Appellants' claim 1.

In general, Subbiah discloses a knowledge-based connection admission method and apparatus for providing efficient multiplexing of data and speech over AAL2. Specifically, Subbiah is directed to asynchronous transfer mode (ATM) networks and, more particularly, a subset of the ATM communications protocols; namely, the ATM adaptation layer 2 (AAL2) environment which provides a fixed length packet transport protocol used for voice communication. Subbiah leverages various features within the ATM network to enable opportunistic insertion of data traffic into speech traffic to replace padding or silence. (Subbiah, Abstract).

Subbiah, however, is entirely unlike Appellants' claim 1. Subbiah fails to teach or even suggest a VoIP gateway or other VoIP teachings, much less determining whether

a destination is serviced by a VoIP gateway, multiplexing first and second voice traffic if the second voice traffic is being provided to the VoIP gateway, or transporting multiplexed voice traffic to the VoIP gateway utilizing a plurality of transport packets in response to an affirmative determination that the destination is serviced by the VoIP gateway, as claimed in Appellants' claim 1.

Therefore, since each of Thornton, Sengodan, and Subbiah fail to teach or suggest the limitations of "determining whether a destination is serviced by a second VoIP gateway; multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," any permissible combination of Thornton, Sengodan, and Subbiah must also fail to teach or suggest such limitations as claimed in Appellants' claim 1. Therefore, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest Appellants' claim 1, as a whole.

As such, Appellants submit that independent claim 1 is patentable over Thornton in view of Sengodan and further in view of Subbiah, and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder. Furthermore, claim 2 depends directly from independent claim 1, and recites additional limitations thereof. As such, and for at least for the same reasons as discussed hereinabove, Appellants submit that this dependent claim is also patentable over Thornton in view of Sengodan and further in view of Subbiah and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder.

Therefore, Appellants respectfully request that this rejection under 35 U.S.C. §103(a) be withdrawn.

Claims 13-14:

As described hereinabove, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest Appellants' claim 1, as a whole. Namely, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest at least the limitations of "determining whether a destination is serviced by a second VoIP

gateway; multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 1.

Appellants' claim 13 includes similar limitations of "a first Voice over Internet Protocol (VoIP) gateway...said first VoIP gateway determining whether said destination is serviced by a second VoIP gateway; said first VoIP gateway multiplexing said first voice traffic with a second voice traffic, if said second voice traffic is being provided to said second VoIP gateway; said first VoIP gateway transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 13. Thus, for at least the reasons discussed hereinabove with respect to claim 1, Appellants submit that Thornton, Sengodan, and Subbiah, alone or in combination, also fail to teach or suggest Appellants' claim 13, as a whole.

As such, for at least the reasons stated above, Appellants respectfully submit that independent claim 13 is not obvious and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder. Furthermore, claim 14 depends directly from independent claim 13, and recites additional limitations thereof. As such, and for at least for the same reasons as discussed hereinabove, Appellants submit that this dependent claim is also patentable over Thornton in view of Sengodan and further in view of Subbiah and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder.

Therefore, Appellants respectfully request that the Examiner's rejections be withdrawn.

Claim 27-28:

As described hereinabove, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest Appellants' claim 1, as a whole. Namely, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest at least the

limitations of "determining whether a destination is serviced by a second VoIP gateway; multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 1.

Appellants' claim 27 includes similar limitations of "means for determining whether said destination is serviced by a second VoIP gateway; means for multiplexing, at said VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and means for transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 27. Thus, for at least the reasons discussed hereinabove with respect to claim 1, Appellants submit that Thornton, Sengodan, and Subbiah, alone or in combination, also fail to teach or suggest Appellants' claim 27, as a whole.

As such, for at least the reasons stated above, Appellants respectfully submit that independent claim 27 is not obvious and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder. Furthermore, claim 28 depends directly from independent claim 27, and recites additional limitations thereof. As such, and for at least for the same reasons as discussed hereinabove, Appellants submit that this dependent claim is also patentable over Thornton in view of Sengodan and further in view of Subbiah and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder.

Therefore, Appellants respectfully request that the Examiner's rejections be withdrawn.

Claim 33:

As described hereinabove, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest Appellants' claim 1, as a whole. Namely, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest at least the

limitations of "determining whether a destination is serviced by a second VoIP gateway; multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 1.

Appellants' claim 33 includes similar limitations of "transporting the voice traffic to a second VoIP gateway utilizing a plurality of transport packets if a destination of the voice traffic is serviced by the second VoIP gateway," as claimed in Appellants' claim 33. Thus, for at least the reasons discussed hereinabove with respect to claim 1, Appellants submit that Thornton, Sengodan, and Subbiah, alone or in combination, also fail to teach or suggest Appellants' claim 33, as a whole.

As such, for at least the reasons stated above, Appellants respectfully submit that independent claim 33 is not obvious and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder.

Therefore, Appellants respectfully request that the Examiner's rejections be withdrawn.

35 U.S.C. §103(a) Rejection of Claims 6-12, 18-26, and 32

The Examiner has rejected claims 6-12, 18-26, and 32 under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent 6,363,065 to Thornton et al. (hereinafter "Thornton") in view of Sengodan et al. and Subbiah and further in view of U.S. Patent 5,600,653 to Chitre et al. (hereinafter "Chitre").

Claim 25:

As described hereinabove, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest Appellants' claim 1, as a whole. Namely, Thornton, Sengodan, and Subbiah, alone or in combination, fail to teach or suggest at least the limitations of "determining whether a destination is serviced by a second VoIP gateway; multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic

if said second voice traffic is being provided to said second VoIP gateway; and transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway," as claimed in Appellants' claim 1.

Appellants' independent claim 25 recites similar relevant limitations to those recited in independent claims 1, 13, 27, and 33. Namely, Appellants' claim 25 includes an apparatus having a processor and a storage device coupled to the processor and including instructions for controlling the processor, where the processor is operative with the instructions to "determine whether said destination is serviced by a second VoIP gateway; multiplex, at said VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and transport said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway."

As such, and at least for the same reasons discussed above with respect to the Examiner's rejection of independent claims 1, 13, 27, and 33, claim 25 is patentable over Thornton, Sengodan and Subbiah and fully satisfies the requirements of 35 U.S.C. §103(a).

Furthermore, Chitre fails to bridge the substantial gap between Thornton, Sengodan, and Subbiah and Appellants' claim 25.

Chitre discloses a technique for improving ATM operation over a communications link having bursty bit errors. Chitre is devoid of any teaching or suggestion of an apparatus adapted to determine whether a destination is serviced by a second VoIP gateway. Chitre is also devoid of any teaching or suggestion of an apparatus adapted to multiplex first voice traffic with a second voice traffic at the VoIP gateway if the second voice traffic is being provided to the second VoIP gateway. Chitre is also devoid of any teaching or suggestion of an apparatus adapted to transport multiplexed voice traffic to a second VoIP gateway utilizing a plurality of transport packets responsive to an affirmative determination that a destination is serviced by the second VoIP gateway.

As such, Appellants submit that independent claim 25 is patentable over Thornton in view of Sengodan and Subbiah and further in view of Chitre and fully

satisfies the requirements of 35 U.S.C. §103. Furthermore, claim 26 depends directly from independent claim 25 and recites additional limitations thereof. As such, and at least for the same reasons as discussed above, Appellants submit that claim 26 is also patentable over Thornton in view of Sengodan and Subbiah and further in view of Chitre and fully satisfies the requirements of 35 U.S.C. §103 and is patentable thereunder.

Therefore, Appellants respectfully request that this rejection under 35 U.S.C. §103(a) be withdrawn.

CONCLUSION

Thus, Appellants submit that none of the claims presently in the application are obvious under the provisions of 35 U.S.C. §103. Consequently, Appellants believe all these claims are presently in condition for allowance.

For the reasons advanced above, Appellants respectfully submit that the rejections of claims 1, 2, 6-14, 18-28, 32 and 33 as being obvious under 35 U.S.C. §103 are improper. Reversal of the rejections of the Final Office Action is respectfully requested.

Respectfully submitted,

1/26/07

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CLAIMS APPENDIX

1. (Previously presented) A method, comprising the steps of:

receiving a first voice traffic at a first Voice over Internet Protocol (VoIP) gateway;

determining whether a destination is serviced by a second VoIP gateway;

multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and

transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet,

wherein said modified RTP packet comprises at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

2. (Previously presented) The method of claim 1, wherein said voice traffic is received within the payload portions of User Datagram Protocol (UDP)/Internet Protocol (IP) packets.

3-5. (Cancelled)

6. (Previously presented) The method of claim 1, wherein said Header Error Check field performs one bit error correction.

7. (Previously presented) The method of claim 1, further comprising the step of communicating messages between said VoIP gateway and said second VoIP gateway.

8. (Original) The method of claim 7, wherein said first VoIP gateway communicates an Open Logical Channel message to said second VoIP gateway including said VoIP gateway's port number and Call Identifier of the calling party.

9. (Original) The method of claim 8, wherein in response to said Open Logical Channel message said second VoIP gateway communicates an Open Logical Channel message including said second VoIP gateway's port number and Call Identifier for the called party.

10. (Original) The method of claim 7, wherein in response to a caller terminating a call said VoIP gateway communicates a Close Logical Channel message including said VoIP gateway's port number and said Call Identifier of the calling party to said second VoIP gateway.

11. (Original) The method of claim 10, wherein in response to said Close Logical Channel message said second VoIP gateway communicates a Close Logical Channel ACK message including said second VoIP gateway's port number and said Call Identifier of the called party.

12. (Original) The method of claim 1, wherein said step of determining is made utilizing a gatekeeper.

13. (Previously presented) In a communication system for transporting voice traffic over an Internet Protocol (IP) network to a destination, apparatus comprising:

a first Voice over Internet Protocol (VoIP) gateway, for receiving a first voice traffic;

said first VoIP gateway determining whether said destination is serviced by a second VoIP gateway;

said first VoIP gateway multiplexing said first voice traffic with a second voice traffic, if said second voice traffic is being provided to said second VoIP gateway;

said first VoIP gateway transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet,

wherein said modified RTP packet comprises at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

14. (Original) The apparatus of claim 13, wherein said voice traffic is received within the payload portions ofUser Datagram Protocol (UDP)/Internet Protocol (IP) packets.

15-17. (Cancelled)

18. (Previously presented) The apparatus of claim 13, wherein said Header Error Check field performs one bit error correction.

19. (Original) The apparatus of claim 18, further comprising the step of communicating messages between said VoIP gateway and said second VoIP gateway.

20. (Original) The apparatus of claim 19, wherein said first VoIP gateway communicates an Open Logical Channel message to said second VoIP gateway including said VoIP gateway's port number and Call Identifier of the calling party.

21. (Original) The apparatus of claim 20, wherein in response to said Open Logical Channel message said second VoIP gateway communicates an Open Logical Channel

message including said second VoIP gateway's port number and Call Identifier for the called party.

22. (Original) The apparatus of claim 21, wherein in response to a caller terminating a call said VoIP gateway communicates a Close Logical Channel message including said VoIP gateway's port number and said Call Identifier of the calling party to said second VoIP gateway.

23. (Original) The apparatus of claim 22, wherein in response to said Close Logical Channel message said second VoIP gateway communicates a Close Logical Channel ACK message including said second VoIP gateway's port number and said Call Identifier of the called party.

24. (Original) The apparatus of claim 13, wherein a gatekeeper is used to determine whether said second VoIP gatekeeper services said destination.

25. (Previously presented) A Voice over Internet Protocol (VoIP) gateway for transporting voice traffic over an Internet Protocol (IP) network to a destination, comprising:

a processor; and

a storage device coupled to said processor and including instructions for controlling said processor, said processor operative with said instructions to:

receive a first voice traffic at said VoIP gateway;

determine whether said destination is serviced by a second VoIP gateway;

multiplex, at said VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and

transport said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein

said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet,

wherein said modified RTP packet comprises at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

26. (Original) A Voice over Internet Protocol (VoIP) gateway for transporting voice traffic over an Internet Protocol (IP) network to a destination as in claim 25, wherein a gatekeeper is used to determine whether said destination is serviced by said second VoIP gateway.

27. (Previously presented) A Voice over Internet Protocol (VoIP) gateway, for transporting voice over an Internet Protocol (IP) network, to a destination, comprising:

means for receiving a first voice traffic at said VoIP gateway;

means for determining whether said destination is serviced by a second VoIP gateway;

means for multiplexing, at said VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and

means for transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet,

wherein said modified RTP packets comprise at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

28. (Original) The VoIP gateway of claim 27, wherein said voice traffic is received within the payload portions ofUser Datagram Protocol (UDP)/Internet Protocol (IP) packets.

29-31. (cancelled)

32. (Previously presented) The VoIP gateway of claim 27, wherein said Header Error Check field performs one bit error correction.

33. (Previously presented) A method, comprising the steps of:

receiving a voice traffic at a first Voice over Internet Protocol (VoIP) gateway;

transporting the voice traffic to a second VoIP gateway utilizing a plurality of transport packets if a destination of the voice traffic is serviced by the second VoIP gateway, wherein the transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport Packet (RTP) packet,

wherein said modified RTP packets comprise at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

34. (cancelled)

EVIDENCE APPENDIX

None

RELATED PROCEEDINGS APPENDIX

None